IN THE UNITED STATES DISTRICT COURT
FOR THE EASTERN DISTRICT OF TEXAS
MARSHALL DIVISION

SAINT LAWRENCE COMMUNICATIONS LLC, **Plaintiff,** **v.** **ZTE CORPORATION, ZTE USA, INC., and ZTE (TX) INC.,** **Defendants.**	**Case No. 2:15-cv-00349-JRG** **(Lead Case)** **JURY TRIAL DEMANDED**
SAINT LAWRENCE COMMUNICATIONS LLC, **Plaintiff,** **v.** **MOTOROLA MOBILITY LLC,** **Defendant.**	**Case No. 2:15-cv-00351-JRG** **(Consolidated Case)** **JURY TRIAL DEMANDED**

DECLARATION OF TOKUNBO OGUNFUNMI, Ph.D.

1. My name is Tokunbo Ogunfunmi. I am offering this declaration in the matter listed above on behalf of Saint Lawrence Communications LLC ("SLC") and at the behest of their attorneys Ahmad, Zavitsanos, Anaipakos, Alavi & Mensing P.C. I am being compensated at my usual rate and my compensation is not dependent on any opinions that I may take in this matter, any testimony, or any intermediate or final resolution in the matter.

2. I have been asked to review the patents in suit in this matter, i.e. U.S. Patents Nos. 6,795,805 (`805 Patent); 6,807,524 (`524 Patent); 7,151,802 (`802 Patent); 7,260,521 (`521 Patent); and 7,191,123 (`123 Patent) (collectively, "the Asserted Patents"). My review of these patents followed my ordinary practice, i.e., I began by reading the patents themselves and then reviewed their respective prosecution histories.[1]

[1] In preparing my declaration I have reviewed the list of claim terms in dispute between the parties as of November 25, 2015 as reflected in the parties' amended Joint Claim Construction Statement. However, it is my understanding that there are ongoing discussions between the parties with respect to the claim terms in dispute and the parties' proposed constructions for those terms. Therefore, I reserve the right to amend and/or supplement my declaration in light of a change in the terms in dispute and/or the parties' positions.

3. My curriculum vitae and testimony list are included in Appendix A to this report. To summarize my qualifications, I hold three academic degrees in the field of Electrical Engineering: A Bachelor of Science and Engineering, a Masters of Science and Engineering, and a Doctorate of Philosophy degree. I received my Bachelor degree (First Class Honors) from University of Ife in Nigeria and my Masters and Doctorate degrees from Stanford University. I have expertise in digital and adaptive signal processing and communications applications of signal processing as well as speech coding. A more comprehensive description of my expertise is reflected in my CV which is attached as Appendix A to this report.

<u>Legal Understanding</u>

4. I am not an attorney. Therefore, I have relied upon certain legal principles that have been explained to me.

5. I have been informed that when interpreting claim terms one must endeavor to employ the perspective of a person of ordinary skill in the relevant art at the time of invention. When interpreting the claims, I understand that the ordinary meaning of the language within the claims should be utilized unless the specification or prosecution history clearly provides reason for applying a different interpretation. In other words if a claim term has a well-known and understood meaning to a person of ordinary skill in the art that meaning should be applied to the claims unless the patentee states a clear and unambiguous alternative meaning that they intend to be in force.

6. I have approached the meaning of the claim terms as one of ordinary skill in the art at the time of the effective filing date of the patents in suit or the patents to which any of them claim priority. This approach puts the priority date for the`123 Patent at November 1999 and the priority date for the remaining Asserted Patents at October 1998. I am currently unaware of any arguments that any of the patents should be allotted an earlier priority date. Therefore, I have no opinion at this moment on any such possible argument.

7. I understand that a number of factors should be considered in determining the level of ordinary skill in the art, including: (1) the educational background of those actively working in the field at the time the invention was made (and particularly of any person(s) who may have independently made the invention(s) at about the same time as the inventor(s)); (2) the type of problems encountered in the art; (3) the various ways that others sought to solve the existing problems; (4) the rapidity with which innovations were being made in the art at the time the invention was made; (5) the level of technological sophistication at the time the invention was made; (6) the educational level of the inventor; and (7) teachings and disclosures of any references that, while not prior art to the invention(s), nonetheless contain teachings or disclosures of what the level of ordinary skill in the field may have been at the time the invention(s).

8. In my opinion, a person of ordinary skill in the art in October 1998 or November 1999 ("relevant time frame") would have had a Bachelor's degree in electrical engineering or computer science and at least 2 years of experience in digital signal processing or speech

processing, or would possess equivalent education or experience. In some cases a greater degree of education could compensate for a lesser degree of work experience. Similarly, in some cases a greater degree of work experience could compensate for a lesser degree of education. When I use the term "person of ordinary skill in the art," I am referring to such a person.

9. I understand that a patent includes a specification that describes the patented invention. The specification includes a written description of the preferred embodiment(s) of the invention, drawings, and claims that define the scope of the patent.

10. I also understand that patents may include dependent and independent claims. Dependent claims include each and every element of the claim or claims from which they depend.

11. I have further been informed that a patent is invalid for indefiniteness if its claims, read in light of the specification delineating the patent, and the prosecution history, fail to inform, with reasonable certainty, those skilled in the art about the scope of the invention.

12. I have been informed that a claim element may be expressed as a means or step for performing a specified function without reciting the specific structure, material or acts in support thereof. I have also been informed that in such instances, the claim element is construed to cover the corresponding structure, material or acts described in the specification and equivalents thereof. I have further been informed that with respect to computer-implemented inventions, the structure disclosed in the specification needs to be more than simply a general purpose computer or microprocessor. Specifically, I have been informed that with respect to such computer-implemented inventions, the specification must disclose an algorithm for performing the claimed function. Further, I have been informed that the patentee may express that procedural algorithm in any understandable terms including as a mathematical formula, in prose, or as a flow chart, or in any other manner that provides sufficient structure.

13. I will attempt to apply these legal principles, as they have been explained to me, in my analysis of the Asserted Patents.

Technology Background

14. The Asserted Patents are directed to improvements to both the decoder and the encoder side of a speech codec which facilitate transmission of wideband speech signals. Specifically, the Asserted Patents are directed to an audio compression model that is particularly effective at modeling human speech. This model (referred to hereinafter as the "Human Speech Model") uses three major components: a fixed codebook, an adaptive codebook, and an LP synthesis filter to model the three main anatomical components of the human speech system: the lungs (1), the larynx (2) and the vocal tract (3) as shown in Figure 1, below.

15. In human speech, the lungs are used to push air up through the trachea. The fixed codebook component essentially models the flow of air coming through the lungs. This air then passes through the larynx which contains the vocal chords. If the sound being made is a voiced sound, such as a vowel—"A" or "E" or "I"—the vocal chords vibrate. If the sound being made is

an unvoiced sound, such as the fricative sound formed by the letters "s" and "h"—"SHHHHH"—the vocal chords do not vibrate. The adaptive codebook component essentially models the periodic vibration of the vocal chords. The flow of air output by the larynx then goes through the vocal tract, where it is amplified and given a specific sound characteristic by the shape of the vocal tract components, such as the position of the tongue and the opening of the lips. The LP synthesis filter component essentially models the shape of the vocal tract. Together, the fixed codebook, the adaptive codebook and the LP synthesis filter form the basis for the audio compression model described in the Asserted Patents.

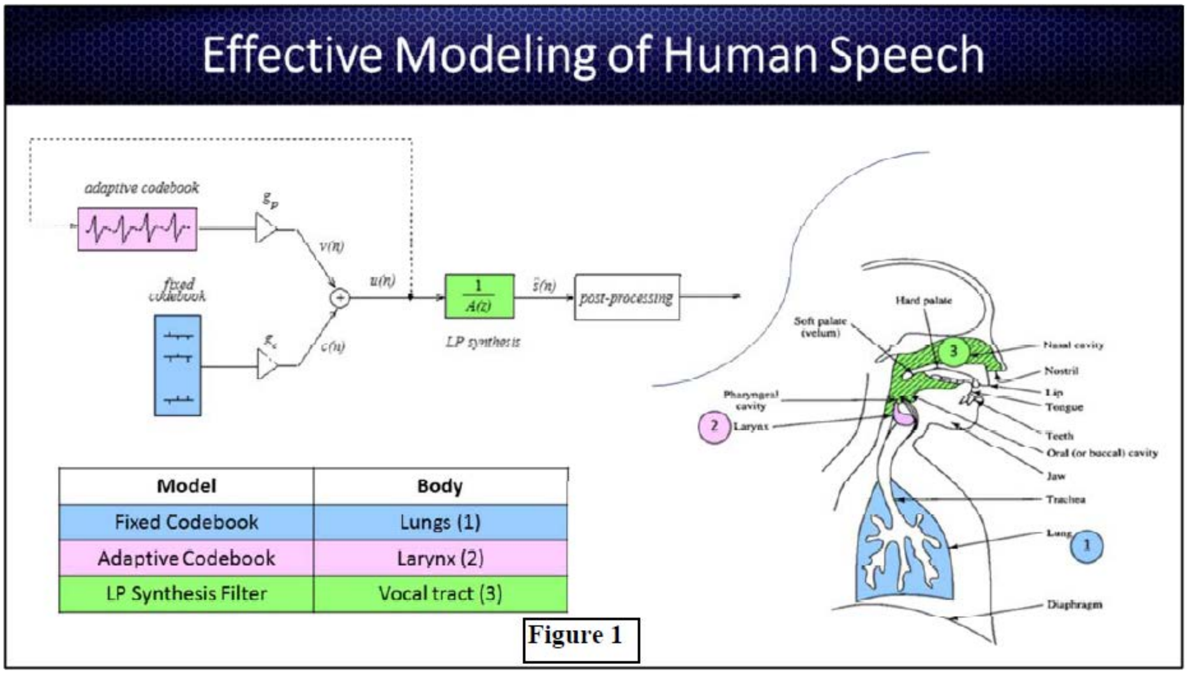

Figure 1

16. Although this model allows an encoder to efficiently model human speech, speech codecs implementing this model face certain problems, particularly in applications involving wideband speech. The Asserted Patents seek to solve these problems.

"Means for" terms

17. The Asserted Patents express a number of claim elements as a "means for" performing a certain function. I understand that there is a dispute between the parties with regard to these terms. Attached as Appendix B is a table which lists the "means for" terms in dispute and with respect to each term, identifies the parties' respective proposed construction.

18. It is my understanding that defendants contend that each of the "means for" terms in Appendix B is indefinite due to a lack of disclosure of a corresponding structure for performing the function recited in the claim. It is my understanding that in contrast, St. Lawrence has identified the corresponding structure for each function as a computer performing a particular algorithm.

19. A person of ordinary skill in the art would understand that the "means for" terms are used in the Asserted Patents in the context of a computer-implemented invention:

a. The Asserted Patents are all generally directed to the compression, transmission, and decompression of digital speech data by mobile devices in a cellular network. [e.g., `123 Pat., 5:31-34 ("Fig. 4 is a simplified, schematic block diagram of a cellular communication system in which the wideband encoder of Fig. 1 and the wideband decoder of Fig. 2 can be used")].
b. The processes described in the Asserted Patents involve complex mathematical computations performed at a rate of thousands of computations per second. For instance, the Asserted Patents note that the audio signals being processed are sampled at 16,000 samples per second. [e.g., `123 Pat., 2:9-11].
c. The Asserted Patents describe performing the compression and decompression algorithms in the digital domain and computer memory is used to store the digital data. [e.g., `123 Pat., 1:24-33 (noting that "[a] speech encoder converts a speech signal into a digital bit stream which is transmitted over a communication channel (or stored in a storage medium)" and that the speech decoder "processes the transmitted or stored bit stream to convert it back to a sound signal").
d. The Asserted Patents use variations of the word "compute" over 200 times, further evidencing the fact that the inventions disclosed therein were implemented using a computer.
e. The Asserted Patents also reference "fixed-point implementation of the algorithm[s] and the use of "single-precision arithmetic," both of which are concepts that are central to digital computing. [`123 Pat., 2:17-18; 8:38-39]. Specifically, "fixed-point" computing refers to the representation of fractional numbers in a computer with a fixed decimal point (e.g., 1.234). In contrast, in "floating-point" computing fractional numbers are represented by a form of scientific notation (e.g., 1234×10^{-3}).
f. It was well known to those of ordinary skill in the art in the relevant time frame that computers were used to encode and/or decode speech data by mathematically processing an audio signal. For example, cellular phones that were publicly available in that time frame used computers (e.g., a processor) in order to encode and/or decode speech signals. Accordingly, speech encoding and decoding operations were routinely performed using computers in the relevant time frame.

20. I have been informed that with respect to such computer-implemented inventions, the specification must disclose an algorithm for performing the claimed function. In this case, with respect to each of the "means for" terms, the specification identifies the relevant algorithm as indicated in Appendix B.

21. Accordingly, a person of ordinary skill in the art would recognize that each "means for" term is implemented using a computer and with respect to each term, the Asserted Patents disclose the specific algorithm that would be implemented by the computer. Thus, from the perspective of a person of ordinary skill in the art in the relevant time frame, the patents disclose specific algorithms performed on a computer that correspond to each of the "means for" terms. Accordingly, a person of ordinary skill in the art would understand the scope of the invention claimed in the Asserted Patents with reasonable certainty.

Meaning of "Wideband [Speech] Signal "

22. The term "wideband" was a term of art in the relevant time frame which was intended to provide a contrast to the term "narrowband" which was used previously to refer to traditional telephone applications that filtered a speech signal in the range of 200-3400 Hz.

23. Due to practical considerations associated with processing speech signals, it was well known to those of ordinary skill in the art that there is not one specific frequency at which a signal goes from being a narrowband speech signal to a wideband speech signal. Instead, there was a general understanding that a wideband speech signal has a bandwidth that is approximately twice as wide as that of a narrowband speech signal.

24. The Asserted Patents acknowledge the demand for "efficient digital wideband speech/audio encoding techniques." [`805 Pat., 1:12-17]. Further, the term "wideband speech" signal was well-known to those of ordinary skill in the art in the relevant time frame and was commonly used as shown, for example, in:

a. P. Mermelstein, "G.722, A new CCITT Coding Standard for Digital Transmission of Wideband Audio Signals," *IEEE* Comm. Mag., Vol. 26, No. 1, pp. 8-15, Jan. 1988 (describing a standard applicable to wideband signals and discussing the frequency range of wideband audio signals compared to narrowband audio signals) (attached as Ex. A);
b. Fuemmeler et. al, "Techniques for the Regeneration of Wideband Speech from Narrowband Speech," EURASIP Journal on Applied Signal Processing 2001:0, 1-9 (Sep. 2001) (noting that some work has already been done in the area of wideband speech regeneration) (attached as Ex. B);
c. C.H. Ritz et. al., "Lossless Wideband Speech Coding," 10th Australian Int'l. Conference on Speech Science & Technology, p. 249 (Dec. 2004) (noting that wideband speech refers to speech sampled at 16 kHz and acknowledging existing research into wideband speech coding) (attached as Ex. C).
d. U.S. 5,581,652, filed Sep. 29, 1993 (titled: "Reconstruction of wideband speech from narrow band speech using codebooks) (Ex. D);
e. U.S. 6,615169, filed Oct. 18, 2000 (titled: "High frequency enhancement layer coding in wideband speech codec") (Ex. E).

25. The AMR-WB Standard implemented by the 3GPP used the term "wideband" in its title further evidencing the fact that the meaning of that term was well-known to those of ordinary skill in the art.

26. The Asserted Patents further make it clear that the wideband [speech] signal discussed therein is not limited to a signal having a bandwidth with a strict cut off at 50-7000 Hz and can include frequency components above 7000 Hz or, alternatively, may not have components all the way up to 7000 Hz:

a. The `805 Patent discloses generating a noise signal in the frequency range of 5.6-7.2 kHz that is then added to the synthesized speech signal to form the wideband speech signal at

the output which would include frequencies in the range of 7000 to 7200 Hz. [`805 Pat., 17:64-18:4].

b. The `802 Patent claims recite a decoder for producing a synthesized wideband signal where the bandpass filter has a frequency bandwidth located between 5.6 kHz and 7.2 kHz. [`802 Pat., 21:10-13].

c. The 802 Patent describes an embodiment where the input wideband signal is down-sampled from 16 kHz to 12.8 kHz, reducing "the number of samples in a frame, the processing time and the signal bandwidth below 7000 Hz." [`802 Pat., 2:48-51; claim 1 ("a wideband signal previously down-sampled during encoding")]. This is due to the sampling theorem according to which the highest frequency component in a signal is equal to half of the sampling rate. Accordingly, if a signal is sampled at 8000 samples per second, the maximum frequency component of the resulting signal will be at 4 kHz. In contrast, when a signal is sampled at 16,000 samples per second, the highest frequency component of the resulting signal will be at 8 kHz. Therefore, for the down-sampled wideband signal discussed in the `802 Patent which is down-sampled from 16 kHz to 12.8 kHz, the highest frequency component would be at 6400 Hz, i.e., below 7000 Hz.

d. The `524 Patent defines the output of the preemphasis filter (103) (i.e., signal "S"), as "the wideband signal input speech vector (after down-sampling, pre-processing, and preemphasis)." [`524 Pat., 7:2-3].The `524 Patent further discloses that in the downsampling module (101) the signal is down-sampled "from 16 kHz down to 12.8 kHz," corresponding to a maximum frequency of 6400 Hz. [`524 Pat., 7:45-48]. Accordingly, the `524 Patent discloses a wideband signal with a frequency range less than 7000 Hz.

27. For the reasons discussed above, a person of ordinary skill in the art in the relevant time frame would understand the meaning of the term wideband [speech] signal. Moreover, a person of ordinary skill in the art in the relevant time frame would understand that a wideband speech signal does not necessarily have to include components of speech in the whole frequency range of 50-7000 Hz. Accordingly, from the perspective of a person of ordinary skill in the art in the relevant time frame, a wideband speech signal may have components that go beyond 7000 Hz or, alternatively, may not have components all the way up to 7000 Hz.

Meaning of "Signal Path"

28. The term "signal path" is widely used and would have been understood by a person of ordinary skill in the art. That the term "signal path" is widely used and commonly understood is evident, for example, from the fact that the same term is used verbatim in the AMR-WB Standard, a standard that is widely implemented by carriers and cellular phone manufacturers. (e.g., Ex. F at p. 25).

29. Independent claim 1 of the `521 Patent recites "at least two signal paths associated to respective set of pitch codebook parameters;" that "each signal path comprises a pitch prediction error calculating device;" and that "at least one of said at least two signal paths comprises a filter for filtering the pitch codevector." [`521 Pat., 18:25-42]. [2] A selector then

[2] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

compares the pitch prediction errors calculated in the different signal paths to choose the signal path having the lowest calculated pitch prediction error and selects the set of pitch codebook parameters associated to the chosen signal path. [`521 Pat., 18:43-48].

30. As discussed with respect to the "means for" terms above, it was known to those of ordinary skill in the art in the relevant time frame that the invention of the Asserted Patents was implemented using computer code in a computer (e.g., a processor). Accordingly, it was well known to those of ordinary skill in the art that such a "signal path" may be implemented as either a physical path for a signal promulgated in a circuit or as a logical path implemented using computer code.

31. I have been informed that in litigation pending in Germany involving the German counterparts of the Asserted Patents, defendants' attempt to limit the meaning of the term "signal path" to a physical path was rejected by the German court which concluded that "the wording of the claim does not restrict 'signal path' to a physical signal path in such a way that it has to be an electronic hardware component and thus a software implementation (in accordance with the standard) is not sufficient. (Ex. G).

32. When implementing the invention of the Asserted Patents using a computer, the "signal path" recited in the claims of the `521 Patent may be implemented as a series of algorithms. Specifically, each "signal path" refers to a logical path that is pursued to identify the set of pitch codebook parameters that result in the lowest calculated pitch prediction error. For instance, a first set of lines of code would be used to implement the first signal path and a second set of lines of code would be used to implement the second signal path when implementing the claimed invention of the `521 Patent. The output of the first set of lines of code and the second set of lines of code would then be compared to identify the set of pitch codebook parameters that result in the lowest calculated pitch prediction error.

33. In fact, the AMR-WB standard implements the very same "signal path" using lines of code, further evidencing the fact that it was well known to those of ordinary skill in the art to use such an implementation. (Ex. F); (Ex. J (excerpts from cod_main.c file).

34. A person of ordinary skill in the art would understand the meaning of the term "signal path." Further, from the perspective of a person of ordinary skill in the art in the relevant time frame a "signal path" is not limited to a physical path for a signal and can be implemented, for example, as a logical signal path using computer code.

Meaning of "Low Frequency Portion"

35. The term "low frequency portion" as used in the claims of the `805 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

36. Independent claim 1[3] of the `805 Patent recites "an innovation filter for filtering the innovative codevector in relation to said periodicity factor to thereby reduce energy of a low

[3] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

frequency portion of the innovative codevector and enhance periodicity of a low frequency portion of the excitation signal."

37. It is common for those of ordinary skill in the art to refer generally to "low frequency portion" and/or "high frequency portion" of a signal without specifying explicit cut off frequencies to delineate where such portions begin or end. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what a "low frequency portion" of a signal is based on the context in which that term is used. This is evident, for example, from:

a. U.S. Pat. No. 8,494,667 (Title: "Apparatus for encoding and decoding audio signal and method thereof") ("In accordance with the present invention, the generation of the ADG is carried out in such a manner that the low frequency portion of the ADG is not generated as a gain, but generated by executing residual coding for the low frequency component of the first downmix signal, and the high frequency portion of the ADG is generated as a gain, as in a conventional method, in order to enable the generated ADG to exhibit an improved performance." at 15:10-17) (attached as Ex. H);
b. U.S. Pat. No. 7,991,495 (Title: "Method and apparatus for processing an audio signal") ("Yet, although the decoding of the extension signal is executed, the decoding can be performed on a predetermined low frequency portion of the extension signal only (1450). For instance, there is a case that since the decoding apparatus is a low power decoder, if the extension signal is entirely decoded, efficiency is degraded, or since the decoding apparatus is unable to decode the entire extension signal a predetermined low frequency portion of the extension signal is usable." 12:61-13:2) (attached as Ex. I);

38. In the context of the `805 Patent, the term "low frequency portion" is used in relation to the output of the innovation filter and would have been understood by a person of ordinary skill in the art.

39. One limitation of the Human Speech Model results from a combination of two factors: (1) audible frequencies in wideband speech extend to very low values (e.g., down to 50 Hz) where the speech signal has high spectral dynamics and the hearing system is highly sensitive to noise between the harmonics of speech (i.e., the peaks in the spectrum of voice speech); and (2) speech compression using the Human Speech Model is lossy (i.e., noise is introduced in the decoded speech) and although this noise may be dynamically shaped at the encoder, it still extends over the whole audible frequency band of the wideband speech signal (i.e., about 50-7000 Hz).

40. The figure below is an illustrative example of a typical wideband speech signal in a voiced segment in the frequency domain, along with an illustration of the noise introduced by the lossy encoder/decoder pair.

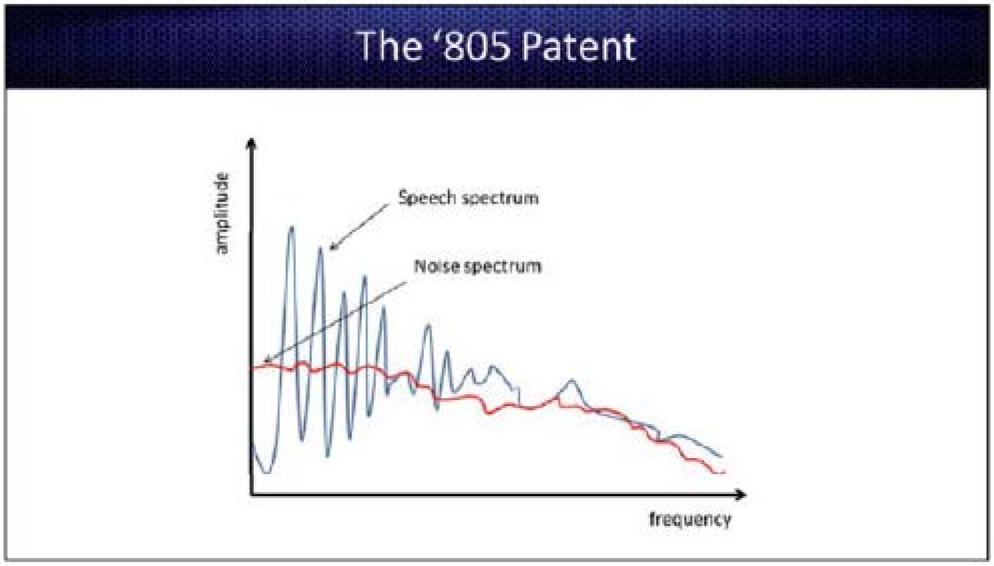

Figure 2

41. As shown in the illustrative figure (Figure 2) above, at the lower frequencies, the relation between the peak of a harmonic and the "valley" between two harmonics is larger compared to the higher frequencies. Accordingly, any noise can have a highly adverse impact in the lower frequency part of the speech spectrum. Moreover, since typically noise levels increase with a decrease in the bit rate, this problem becomes increasingly apparent at lower bit rates. The `805 Patent proposes a solution to this problem.

42. The `805 Patent discloses a solution whereby a decoder at the receiver modifies the content of the fixed codebook as a post processing step. The encoder is not affected by this post processing step which is only applied at the decoder. Since an encoder/decoder pair using the Human Speech Model introduces noise in the signal, this noise is interpreted as originating from the fixed codebook. The noise is due to the analysis-by-synthesis procedure where the decoder is embedded in the encoder. The approximation error resulting from the difference between the "best entry" from the fixed codebook and the actual portion(s) of the speech signal not modeled by the LP synthesis filter and the adaptive codebook is the coding noise.

43. Specifically, at the encoder of the transmitter, the first step is to calculate the parameters of the LP synthesis filter. The second step is to calculate the adaptive codebook (also referred to as "pitch codebook") parameters and the third step is to identify the best entry from the fixed codebook (also referred to as "innovative codebook"). The identification of the best entry from the fixed codebook is an approximation in the sense that it aims to identify the entry which best models, without being equal to, the portion(s) of the speech signal that the LP synthesis filter and the adaptive codebook were not able to model. The approximation error resulting from the difference between the "best entry" from the fixed codebook and the actual portion(s) of the speech signal not modeled by the LP synthesis filter and the adaptive codebook is the coding noise.

44. One way to reduce the coding noise is to have an encoder that makes a better approximation when identifying the "best entry" from the fixed codebook. However, the availability of a limited number of bits (depending on the bit rate) limits the encoder's ability to improve this approximation. Another solution is to increase the bit rate, which effectively, increases the number of available possibilities for the "best entry" in the fixed codebook. However, the bit rate available to the encoder is constrained by the transmission network.

45. The `805 Patent proposes a third solution which decreases the perception of the coding noise in a post-processing step at the decoder. To that end, the low frequency content of the fixed codebook is reduced in proportion to the periodicity of the decoded signal, i.e., the higher the periodicity of the decoded signal, the more attenuation is applied to the low frequencies of the fixed codebook. Although the proposed solution reduces the optimality of the fixed codebook entry selected by the encoder, this reduction in optimality only occurs in the low frequencies and only when the decoded signal is mostly periodic. Therefore, the solution proposed by the `805 Patent significantly reduces the possibility of perceiving unwanted coding noise in the low frequencies without a significant impact on the decoded signal in any other frequency band.

46. Accordingly, the `805 Patent discloses calculating a periodicity factor for the wideband speech signal at the decoder (i.e., the excitation signal "u" in the `805 Patent) and filtering the selected entry (innovative codevector) from the fixed codebook (i.e., innovative codebook) through an innovation filter (205 -- F(z)) whose coefficients are calculated according to the periodicity factor of the wideband speech signal at the decoder.

47. First, a voicing factor rv is computed in the voicing factor generator (204) as follows:

$$\boldsymbol{rv} = \frac{\boldsymbol{Ev} - \boldsymbol{Ec}}{\boldsymbol{Ev} + \boldsymbol{Ec}}$$

where Ev is the energy of the scaled pitch codevector bvT (i.e., the decoded, scaled codevector v(n) from the adaptive codebook of the Human Speech Model); and Ec is the energy of scaled innovative codevector gck (i.e., the decoded, scaled codevector c(n) from the fixed codebook of the Human Speech Model). Accordingly, if the decoded signal exhibits a speech like periodic nature (i.e., if Ev is much larger than Ec) the ratio rv will be close to 1 and if the decoded signal is noise like (i.e., if Ec is much larger than Ev) the ratio rv will be close to -1. The ratio rv is between -1 and +1.

48. A periodicity factor (α) is then calculated as:

$$\alpha = 0.125(1 + rv)$$

49. Accordingly, α is close to 0.25 if the decoded speech signal exhibits a speech like periodic nature and α is close to 0 if the decoded signal is noise-like and not periodic.

50. An innovation filter is then constructed as a linear, 3-tap, symmetrical finite impulse response (FIR) filter with the transfer function:

$$F(z) = -\alpha z + 1 - \alpha z^{-1}$$

51. The frequency response of the innovation filter (205 – F(z)) is therefore shaped based on the periodicity factor and emphasizes the higher frequencies more than the lower frequencies. Since α is positive or equal to 0, the innovation filter F(z) is always a high pass filter (i.e., attenuates low frequency components and lets the higher frequency components pass through), with a stronger attenuation of the low frequencies when α is largest (i.e., when the decoded wideband speech signal is more periodic or voiced).

52. The selected entry (innovative codevector) from the fixed codebook (i.e., innovative codebook) is filtered through F(z). The coefficients of the innovation filter (F(z)) are calculated such that this filter will reduce the energy of the low frequency portion of the innovative codevector (the entry selected from the fixed codebook) and thereby enhance the periodicity of the total excitation signal (i.e., sum of the contributions from the fixed codebook and the adaptive codebook). This reduction in the intensity of the coding noise in the low frequencies is shown in the illustrative figure below (Figure 3):

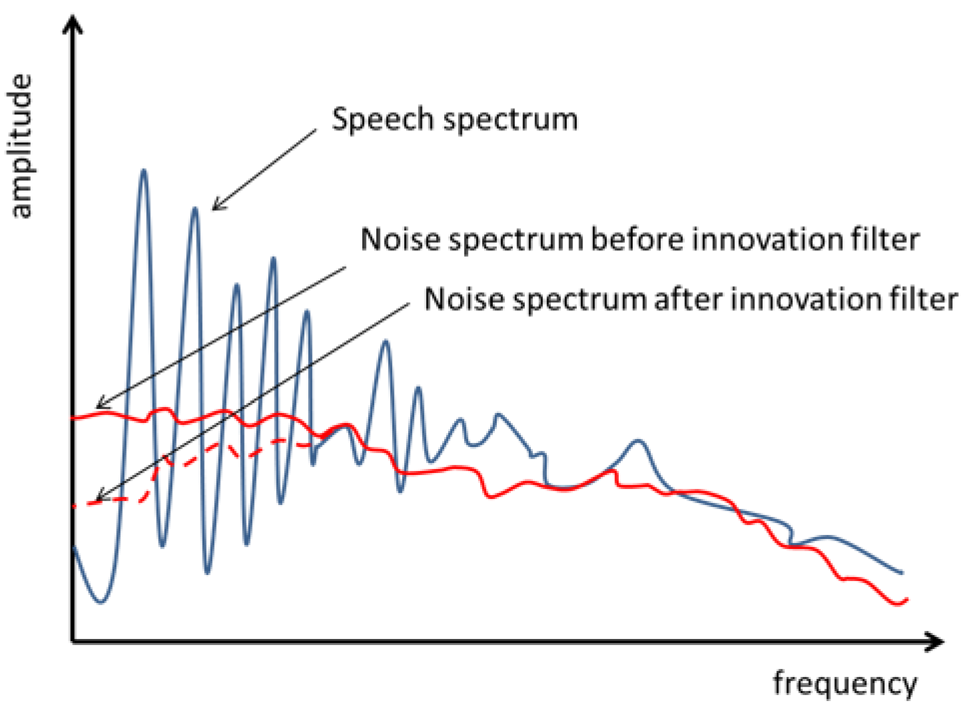

Figure 3

53. Accordingly, the proposed solution of the `805 Patent reduces the perceived distortion at the receiver and improves the quality for the listener. Moreover, most of the energy of the harmonics (peaks in the speech spectrum) in voiced speech comes from the adaptive codebook. Therefore, the innovation filter (F(z)) will not have much impact on the harmonic

peaks of the speech signal. Instead, as shown in the figure above, the innovation filter (F(z)) mostly affects (i.e., reduces) the noise level at low frequencies.

54. The term "low frequency portion" is used to describe the result of applying the innovation filter. Further, in the context of the invention disclosed in the `805 Patent (as described above), from the perspective of a person of ordinary skill in the art in the relevant time frame the meaning of the term "low frequency portion" and the claim scope would have been understood with reasonable certainty.

Meaning of "Generally Higher"

55. The term "generally higher" as used in the claims of the `802 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

56. Independent claim 1[4] of the `802 Patent recites "a spectral shaper for filtering said scaled white noise sequence in relation to a bandwidth expanded version of the linear prediction filter coefficients to produce a filtered scaled white noise sequence characterized by a frequency bandwidth generally higher than a frequency bandwidth of said over-sampled synthesized signal version."

57. It is common for those of ordinary skill in the art to use relative terms when describing certain operations involving speech signals. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what it means for a white noise sequence to be characterized by having a frequency bandwidth that is "generally higher" than a frequency bandwidth of the over-sampled synthesized signal version.

58. In the context of the `802 Patent, the term "generally higher" is used in relation to the output of spectral shaper and would have been understood by a person of ordinary skill in the art.

59. One limitation of the Human Speech Model is that it does not permit an explicit control of bit allocation over the different frequency bands of the wideband speech signal. Specifically, most of the signal energy in a wideband speech signal lies in the first few thousand Hertz. This is illustrated in the figure below which is shown for illustrative purposes.

---

[4] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which the term "signal path" is in dispute.

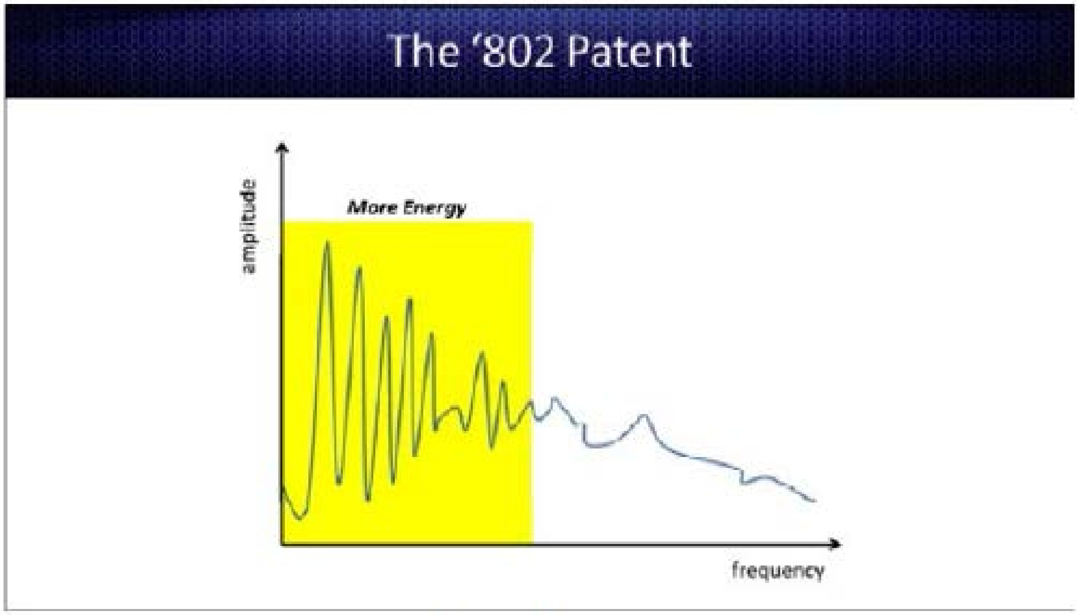

Figure 4

60. Moreover, human perception is more sensitive at low frequencies than at higher frequencies. Considering the fact that there are a limited number of bits available for representing a speech signal, it is beneficial to dedicate more bits to the lower frequencies. Because the model discussed above operates in the time domain, it cannot manipulate how bits are spent over different frequency bands.

61. The `802 Patent proposes using a weighting filter W(z) at the encoder to help control how the coding accuracy is distributed across the signal frequency spectrum, although it is not possible to completely exclude some frequency bands from the encoding process. Accordingly, the solution proposed by the `802 Patent is particularly useful at lower bit rates to allow the limited bit budget to be concentrated in the lower frequencies where the important features of the speech signal lie.

62. The `802 Patent proposes a solution which takes advantage of imperfections in human perception and operates the human speech model in a down-sampled domain. The disclosed solution includes a downsampling module (101) at the encoder side and an oversampler (209) at the decoder side.

63. At the encoder, the input wideband speech originally sampled at 16,000 samples per second is down-sampled at the downsampling module (101) to 12,800 samples per second (i.e., factor of 4/5 down-sampling) prior to being encoded by the encoder. Per the sampling theorem, anytime a continuous (analog) signal is sampled at a sampling rate Fs, it cannot have a bandwidth larger than Fs/2. Accordingly, the original wideband speech sampled at 16,000 samples per second is limited to an upper frequency of 8000 Hz. In contrast, the down-sampled wideband speech is limited to an upper frequency of 6400 Hz (i.e., 12800/2). As a result, the downsampling process removes some of the upper frequency components of the input wideband

speech signal. The encoder then operates on this down-sampled signal and generates a compressed bitstream which is transmitted to a receiver. By operating on the down-sampled signal, the encoder is able to utilize the available bits in the lower frequencies which are of higher importance to the speech signal.

64. The compressed bitstream from the transmitter is received at the receiver. The decoder at the receiver takes this compressed bitstream as an input and generates a synthesized speech in the down-sampled domain ("down-sampled synthesized speech"). The decoder at the receiver is aware that the original wideband speech signal was down-sampled at the encoder. Accordingly, at the oversampler (209), the decoder upsamples the down-sampled synthesized speech signal to generate a synthesized speech signal at the right sampling frequency for wideband speech (i.e., 16,000 samples per second). However, since up-sampling does not reintroduce the high frequency components that were lost during the down-sampling process, the oversampled signal has missing content at those high frequencies.

65. Since no bits were used at the encoder to describe the high frequency components (i.e., components above 6400 Hz) of the input wideband speech signal, the receiver must generate those components. Accordingly, a random noise generator (213) at the receiver generates a random noise signal (sampled at 16,000 samples per second), unrelated to the missing high frequency components of the original speech signal. The noise signal generated is scaled according to the spectral tilt in the decoded speech at the gain adjusting module (214). For instance, when there is more spectral tilt (i.e., a less flat spectrum with more energy in low frequencies), the noise energy is reduced so there is less contribution in high frequencies. A spectral shaper (215) then processes the scaled noise signal using the coefficients of the LP synthesis filter available at the decoder. Specifically, the spectral shaping of the scaled noise at the spectral shaper (215) involves a filtering operation of the noise signal (sampled at 16,000 samples per second) by the LP synthesis filter (206) (calculated on the 12,800 samples per second signal). The result is bandpass filtered by the filter (216) passing the signal in the range from 5600-7200 Hz and then added by the adder (221) to the oversampled synthesized signal ($\hat{S}$) from the oversampler (209) with the missing high frequencies. Accordingly, the signal (Sout) at the output of the adder (221) is a synthesized speech signal at 16,000 samples per second with a full band (up to 7200 Hz) of signal content.

66. Although the signal content above 6400 Hz is not correlated to the original input wideband speech, the scaling at the gain adjusting module (214) and the spectral shaping at the spectral shaper (215) result in a signal that is perceived by a user as being similar to the original speech signal.

67. Moreover, the `802 Patent discloses that the spectral shaper produces the filtered scaled white noise "by filtering the noise Wg through a bandwidth expanded version of the same LP synthesis filter used in the down-sampled domain ($1/\hat{A}(z/0.8)$)." [`802 Pat., 19:30-34]. A person of ordinary skill in the art would understand the scope of the recited "generally higher" term in light of this disclosed transfer function.

68. Accordingly, the `802 Patent uses a high frequency content generation at the decoder to restore the full band of the original signal. [`802 Pat., 17:57-18:3]. Because the scaled white noise is generated and used to fill the upper part of the spectrum that is missing from the original signal, a person of ordinary skill in the art would readily understand that the filtered scaled white noise has a "generally higher" bandwidth than the over-sampled synthesized signal.

69. The term "generally higher" is used to describe the output of the spectral shaper. Further, in the context of the invention disclosed in the `802 Patent (as described above), from the perspective of a person of ordinary skill in the art in the relevant time frame the meaning of the term "generally higher" and the claim scope would have been understood with reasonable certainty.

Meaning of "Substantially Decoupled"

70. The term "substantially decoupled" as used in the claims of the `524 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

71. Independent claim 1[5] of the `524 Patent recites "a perceptual weighting filter, responsive to said preemphasised signal and said synthesis filter coefficients, for filtering said preemphasised signal in relation to said synthesis filter coefficients to thereby produce said perceptually weighted signal, said perceptual weighting filter having a transfer function with fixed denominator whereby weighting of said wideband speech signal in a formant region is substantially decoupled from a spectral tilt of said wideband speech signal."

72. It is common for those of ordinary skill in the art to use relative terms when describing certain operations involving speech signals. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what it means for weighting of the wideband speech signal in a formant region to be substantially decoupled from a spectral tilt of said wideband speech signal.

73. In the context of the `524 Patent, the term "substantially decoupled" is used to describe a characteristic of the perceptually weighted signal that is output by the perceptual weighting filter and would have been understood by a person of ordinary skill in the art in the relevant time frame.

74. The operation of the encoder/decoder of the Human Speech Model is a lossy operation with distortion introduced in the synthesized speech produced by the decoder. The encoder attempts to minimize this distortion by selecting the best set of values for the parameters of the Human Speech Model – i.e., the LP synthesis filter, the adaptive codebook and the fixed codebook. The selection of the optimal values is done using the analysis by synthesis principle.

[5] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which the term "signal path" is in dispute.

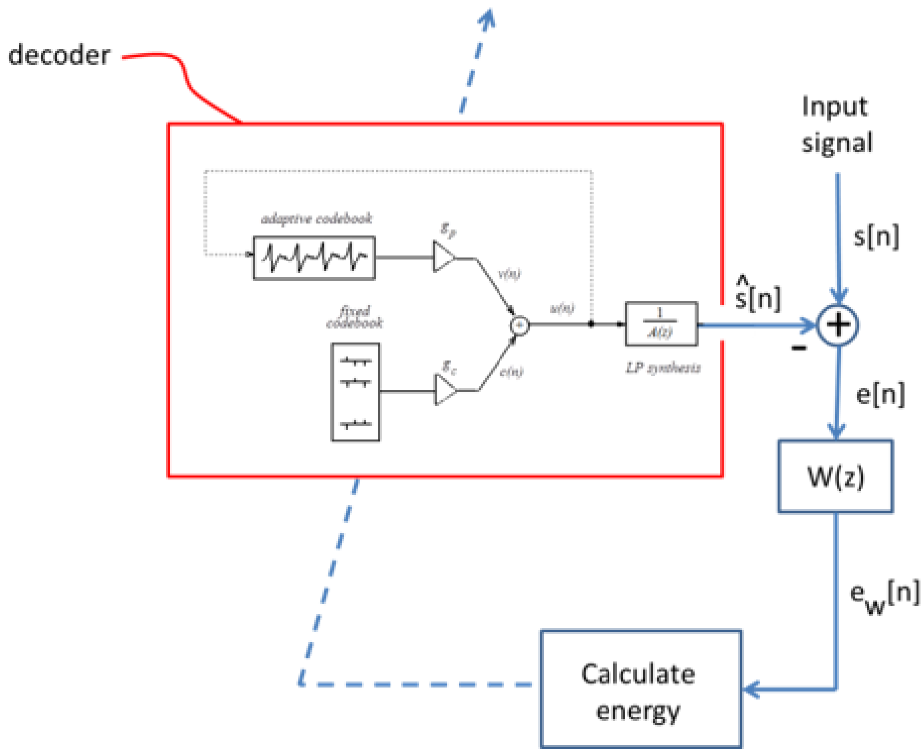

Figure 5

75. As shown in the figure above (Figure 5), the encoder incorporates the decoder. For a given set of parameters for the LP synthesis filter, the adaptive codebook and the fixed codebook (collectively, "Speech Encoding Parameters"), the decoder will generate a given block of synthesized speech ŝ[n]. Accordingly, the encoder searches for the best set of Speech Encoding Parameters which minimize the error between the block of input speech s[n] and the block of synthesized speech ŝ[n]. This analysis-by-synthesis loop is shown in Figure 5 above whereby the encoder tries several possibilities for the Speech Encoding Parameters in the decoder model before identifying the best set of Speech Encoding Parameters for transmission from the transmitter to the receiver.

76. Because the representation of a speech signal (s[n]) using Speech Encoding Parameters is a lossy process, the error (e[n]) which is indicative of the difference between the input speech s[n] and the synthesized speech ŝ[n] will not be 0. However, in order to minimize the perceived effect of the distortion, the encoder uses a weighting filter W(z) to obtain a weighted error $e_w[n]$ and attempts to minimize the energy of this weighted error $e_w[n]$. Accordingly, when the optimal Speech Encoding Parameters are identified, the weighted error $e_w[n]$ will be as small as possible and as close to white noise as possible. With all signals and filters indicated in the transform domain, the relationship between the input speech s[n] and the synthesized speech ŝ[n] in the system shown in the figure above is as follows:

$$\hat{S}(z) = S(z) - W^{-1}(z)E_w(z)$$

77. Since the coding noise $E_w(z)$ has a substantially flat spectrum, the shape of the noise introduced by the encoder is essentially defined by the inverse transfer function $W^{-1}(z)$. The effect of the perceptual weighting filter (W(z)) on the coding noise is illustrated in Figure 6 below. As shown below, the noise spectrum is shaped to mimic the speech spectrum, thereby allowing the speech to mask the noise and reducing the noise perceived by a user.

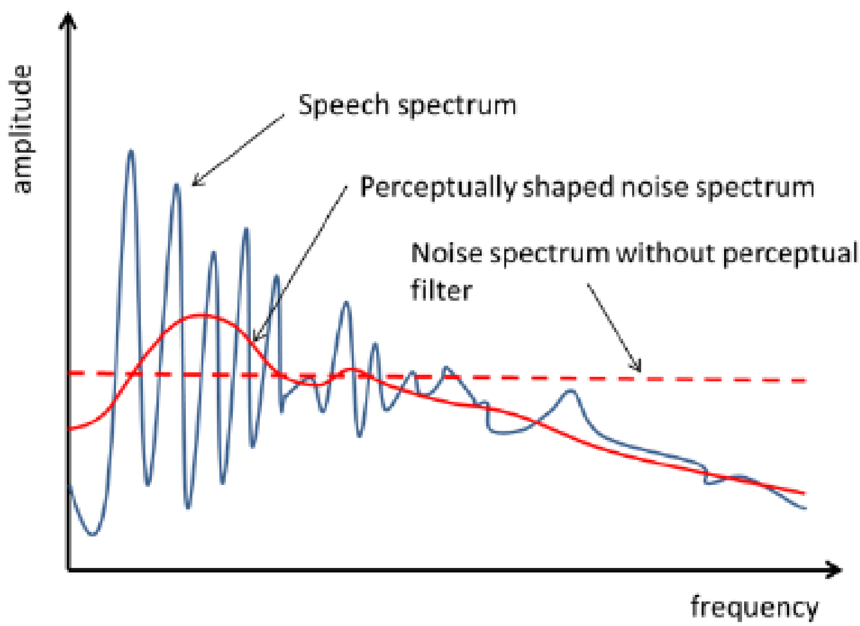

Figure 6

78. Typically, the weighting filter used at the encoder is defined as follows where γ_2 is smaller than γ_1 and both are between 0 and 1:

$$W(z) = \frac{A(\frac{z}{\gamma_1})}{A(\frac{z}{\gamma_2})}$$

79. This weighting filter implies two filtering operations of the order of the LP synthesis filter.

80. Accordingly, the noise weighting filter is derived from the LP synthesis filter which models the speech spectrum envelope. This classical form of the weighting filter W(z) ensures that the noise shape (i.e., $W^{-1}(z)$) follows the energy distribution of the speech spectrum without too much emphasis on the noise in low frequencies if suitable values of γ_1 and γ_2 are selected.

81. Additionally, the spectrum of a wideband speech signal typically exhibits much larger spectral dynamics (i.e., ratio of maximum value to minimum value in the spectrum)

compared to a narrowband speech signal. Figure 7 below is an example to illustrate the difference in spectral dynamics of a wideband speech signal and a narrowband speech signal:

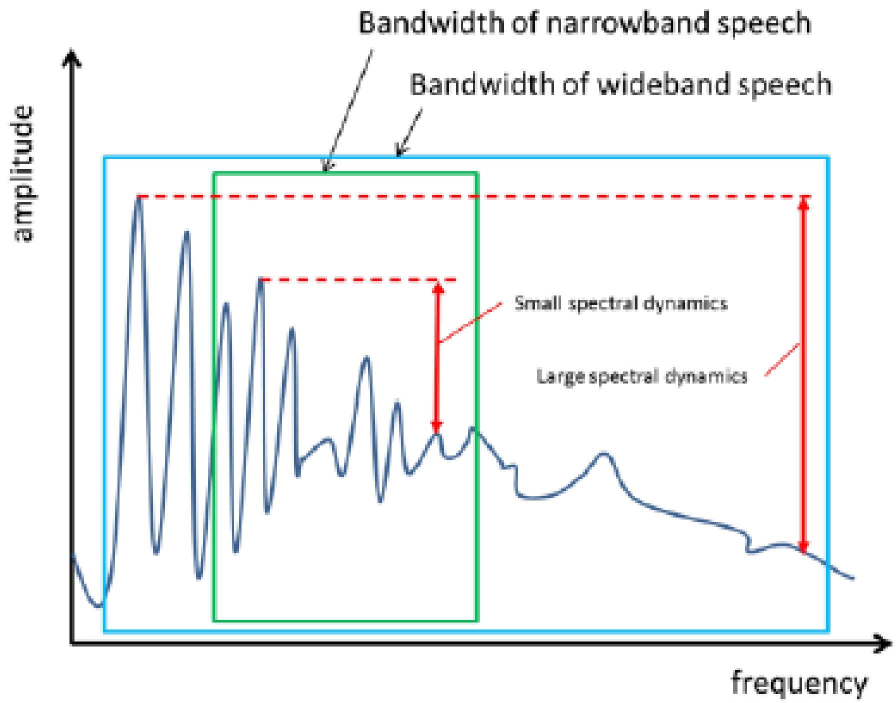

Figure 7

82. Accordingly, when preprocessing a wideband speech signal to be encoded, it is desirable to reduce the spectral dynamics of the signal, in particular for fixed point implementations (e.g., in cellular phones) where a limited resolution (i.e., number of bits) is permitted for representing the data. In signals with large spectral dynamics, implementation of arithmetic with limited resolution can result in inefficient calculations, especially for the LP synthesis filter coefficients. The encoder thus operates in a pre-emphasis domain and a de-emphasis filter must be applied at the decoder to inverse this effect. Therefore, the de-emphasis filter (D(z)) must be included in the overall noise shaping process as follows:

$$\hat{S}(z) = S(z) - W^{-1}(z)D(z)E_w(z)$$

83. As a result, in order to properly encode a wideband speech signal applying both the typical noise shaping filter of the form:

$$W(z) = \frac{A\left(\frac{z}{\gamma_1}\right)}{A\left(\frac{z}{\gamma_2}\right)}$$

84. And the de-emphasis filter D(z), the overall noise shaping is governed by:

$$W^{-1}(z)D(z)$$

85. Therefore, the effect of the de-emphasis filter (D(z)) is explicitly included in the noise introduced in the decoded signal. Since the pre-emphasis filter emphasizes the high frequencies (to reduce spectral dynamics), the de-emphasis filter re-emphasizes the low frequencies at the decoder. As a result, the de-emphasis filter at the decoder effectively pushes the noise upwards at lower frequencies. Accordingly, the prior art approach was unsuitable for wideband applications because of difficulties in modelling the formant structure and the required spectral tilt concurrently. [`524 Pat., 9:20-31].

86. The `524 Patent proposes a solution which reduces the complexity of the applied filter and ensures that the noise shaping process does not introduce too much distortion in the low frequencies.

87. A pre-emphasis filter (103) is introduced at the input of the encoder, the LP filter (A(z)) is computed based on the pre-emphasized speech s(n), and a modified weighting filter (W(z)) having a fixed denominator is used. Specifically, the improved encoder disclosed in the `524 Patent applies the following steps to each block of wideband speech signal that is to be encoded:

(a) A pre-emphasis filter (103) is applied to enhance the high frequency contents and reduce the dynamic range (i.e., reduce spectral dynamics) of the input signal. The pre-emphasis filter has the following transfer function:

$$P(z) = 1 - \mu z^{-1}$$

where μ is between 0 and 1.

(b) The coefficients of the LP synthesis filter (A(z)) are then determined in the calculator module (104) based on the pre-emphasized speech signal (S) from the pre-emphasis filter (103).

(c) A new perceptual weighting filter (105) with a fixed denominator is applied to the pre-emphasized signal from the pre-emphasis filter (103). The new perceptual weighting filter (105) has a transfer function as follows:

$$W(z) = \frac{A\left(\frac{z}{\gamma_1}\right)}{1 - \gamma_2 z^{-1}}$$

(d) The transfer function W(z) has a fixed denominator (independent of the LP filter A(z)). Therefore, the transfer function (W(z)) of the perceptual weighting filter (105) substantially decouples the formant weighting from the spectral tilt of the signal. There is not a complete decoupling of formant weighting from the spectral tilt of the signal because the numerator is still dependent on the LP filter A(z).

(e) The de-emphasis filter on the decoder side (D(z)) is defined as:

$$D(z) = \frac{1}{P(z)}$$

(f) Accordingly, with $\gamma_2 = \mu$, the total coding noise will have the shape:

$$W^{-1}(z)D(z) = \frac{A^{-1}\left(\frac{z}{\gamma_1}\right)}{P^{-1}(z)}D(z)$$

(g) Because of the relationship between D(z) and P(z), the total coding noise is:

$$W^{-1}(z)D(z) = A^{-1}\left(\frac{z}{\gamma_1}\right)D^{-1}(z)D(z) = A^{-1}\left(\frac{z}{\gamma_1}\right) = \frac{1}{A(\frac{z}{\gamma_1})}$$

(h) Because the fixed denominator of the perceptual weighting filter is equal to the pre-emphasis filter, it cancels out with the de-emphasis filter when considering the noise shaping of the encoder/decoder pair.

88. Accordingly, the improved solution proposed by the `524 Patent results in a decoupling of the de-emphasis filter (D(z)) from the coding noise shaping. Additionally, it reduces the complexity of the perceptual weighting filter W(z) since it uses the shape: $W(z) = \frac{A(z/\gamma_1)}{1-\mu z^{-1}}$ (16^{th} order at numerator and 1^{st} order at denominator) instead of $W(z) = \frac{A(z/\gamma_1)}{A(z/\gamma_2)}$ (16^{th} order at numerator and 16^{th} order at denominator). Further, since the LP filter A(z) is calculated on the pre-emphasized speech signal, the spectral shape of the total coding noise $\frac{1}{A(\frac{z}{\gamma_1})}$ has less predominance in the low frequencies which properly shapes the coding noise as shown in the illustrative example of Figure 6, *supra*.

89. Accordingly, the solution proposed by the `524 Patent is to "introduce the pre-emphasis filter 103 at the input, compute the LP filter A(z) based on the pre-emphasized speech s(n), and use a modified filter W(z) by fixing its denominator." [`524 Pat., 9:32-36]. To that end, the `524 Patent discloses "a new perceptual weighting filter 105 with [a] fixed denominator." [`524 Pat., 9:32-36]. It is the application of this improved perceptual weighting filter which "substantially decouples the formant weighting from the tilt." [`524, 9:44-45].

90. Dependent claim 2, which depends on independent claim 1 defines a transfer function for the signal pre-emphasis filter of independent claim 1. [`524 Pat., 18:45-51]. Further, dependent claim 4 which depends on dependent claim 2 (and independent claim 1) defines a transfer function for the perceptual weighting filter of independent claim 1.[`524 Pat., 18:54-61]. In view of the transfer functions defined in dependent claims 2 and 4, a person of ordinary skill in the art would understand the meaning of the term "substantially decouple" and the scope of the claim with reasonable certainty.

91. The term "substantially decoupled" is used to describe the effect of applying the disclosed transfer functions of the signal pre-emphasis filter and the perceptual weighting filter. Further, in the context of the invention disclosed in the `524 Patent (as described above), from the perspective of a person of ordinary skill in the art in the relevant time frame the meaning of the term "substantially decoupled" and the claim scope would have been understood with reasonable certainty.

Meaning of "Reduce a Difference Between the Wideband Speech Signal and a Subsequently Synthesized Wideband Speech Signal"

92. The term "reduce a difference between a wideband speech signal and a subsequently synthesized wideband speech signal" as used in the claims of the `524 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

93. Independent claim 1[6] of the `524 Patent recites "[a] perceptual weighting device for producing a perceptually weighted signal in response to a wideband speech signal in order to reduce a difference between the wideband speech signal and a subsequently synthesized wideband speech signal."

94. The operation of the encoder/decoder of the Human Speech Model is a lossy operation with distortion introduced in the synthesized speech produced by the decoder. The encoder attempts to minimize this distortion by selecting the best set of values for the parameters of the Human Speech Model – i.e., the LP synthesis filter, the adaptive codebook and the fixed codebook. The selection of the optimal values is done using the analysis by synthesis principle.

95. As shown in Figure 5 (above), the encoder incorporates the decoder. For a given set of parameters for the LP synthesis filter, the adaptive codebook and the fixed codebook (collectively, "Speech Encoding Parameters"), the decoder will generate a given block of synthesized speech $\hat{s}[n]$. Accordingly, the encoder searches for the best set of Speech Encoding Parameters which minimize the error between the block of input speech s[n] and the block of synthesized speech $\hat{s}[n]$. This analysis-by-synthesis loop is shown in the figure above whereby the encoder tries several possibilities for the Speech Encoding Parameters in the decoder model before identifying the best set of Speech Encoding Parameters for transmission from the transmitter to the receiver.

96. Accordingly, from the perspective of one of ordinary skill in the art, the term "reduce a difference between a wideband speech signal and a subsequently synthesized wideband speech signal" refers to the process disclosed in the `524 Patent whereby encoding parameters are selected which minimize the error between the block of input speech s[n] and the block of synthesized speech $\hat{s}[n]$. Minimization of this error (i.e., the difference between a wideband speech signal and a subsequently synthesized wideband speech signal) results in a synthesized wideband speech signal that more closely resembles the original input wideband speech signal.

Meaning of "High Frequency Content"

97. The term "high frequency content" as used in the claims of the `524 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

[6] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

98. Independent claim 1[7] of the `524 Patent recites "a signal pre-emphasis filter responsive to the wideband speech signal for enhancing a high frequency content of the wideband speech signal to thereby produce a pre-emphasized signal."

99. It is common for those of ordinary skill in the art to refer generally to "low frequency portion" and/or "high frequency portion" of a signal without specifying explicit cut off frequencies to delineate where such portions begin or end. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what a "high frequency content" of a signal is based on the context in which that term is used. This is evident, for example, from:

a. U.S. Pat. No. 8,494,667 (Title: "Apparatus for encoding and decoding audio signal and method thereof") ("In accordance with the present invention, the generation of the ADG is carried out in such a manner that the ***low frequency portion*** of the ADG is not generated as a gain, but generated by executing residual coding for the low frequency component of the first downmix signal, and the ***high frequency portion*** of the ADG is generated as a gain, as in a conventional method, in order to enable the generated ADG to exhibit an improved performance." at 15:10-17) (attached as Ex. H);
b. U.S. Pat. No. 7,991,495 (Title: "Method and apparatus for processing an audio signal") ("Yet, although the decoding of the extension signal is executed, the decoding can be performed on a predetermined ***low frequency portion*** of the extension signal only (1450). For instance, there is a case that since the decoding apparatus is a low power decoder, if the extension signal is entirely decoded, efficiency is degraded, or since the decoding apparatus is unable to decode the entire extension signal a predetermined ***low frequency portion*** of the extension signal is usable." 12:61-13:2) (attached as Ex. I);

100. In the context of the `524 Patent, the term "high frequency content" is used in relation to the output of the innovation filter and would have been understood by a person of ordinary skill in the art.

101. The function of the pre-emphasis filter (103) is to enhance the high frequency contents of the input speech signal. [`524 Pat., 8:9-10]. Specifically, as discussed above, the spectrum of a wideband speech signal typically exhibits much larger spectral dynamics (i.e., ratio of maximum value to minimum value in the spectrum) compared to a narrowband speech signal. Figure 7 (above) is an example to illustrate the difference in spectral dynamics of a wideband speech signal and a narrowband speech signal.

102. Accordingly, when preprocessing a wideband speech signal to be encoded, it is desirable to reduce the spectral dynamics of the signal, in particular for fixed point implementations (e.g., in cellular phones) where a limited resolution (i.e., number of bits) is permitted for representing the data. In signals with large spectral dynamics, implementation of arithmetic with limited resolution can result in inefficient calculations, especially for the LP synthesis filter coefficients. The encoder thus operates in a pre-emphasis domain by enhancing

[7] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

the high frequency content of the input speech signal which results in a reduction in the dynamic range of the input speech signal. [`524 Pat., 8:9-14].

103. One of ordinary skill in the art would understand that the specific frequency range for what constitutes this "high frequency content" is not necessary to understand the claim scope with reasonable certainty. Instead, the term "high frequency content" is intended as a term to describe this content in contrast to the low frequency content of the speech signal which typically has a higher amplitude. In fact, the specific cut off frequency for the "high frequency content" relative to a low frequency content is in part, a function of the particular speech sound being transmitted. Therefore, one of ordinary skill in the art would understand that it would not be reasonable to define the term "high frequency content" in terms of specific frequency cut offs. Instead, one of ordinary skill in the art would understand that the use of a relative term, i.e., "high frequency content" is appropriate to describe the claimed invention of the `524 Patent.

Meaning of "Means for Amplifying"

104. The term "means for amplifying the found codevector with said smoothing gain to thereby produce said gain-smoothed codevector" as used in the claims of the `123 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

105. Independent claim 21[8] of the `123 Patent recites means for amplifying the found codevector with said smoothing gain to thereby produce said gain-smoothed codevector." It is my understanding that defendants are taking the position that this term is indefinite. In contrast, it is my understanding that St. Lawrence has identified: (a) amplifying the found codevector with said smoothing gain to thereby produce said gain-smoothed codevector" as the corresponding function; and (b) the amplifier (232) in figure 2 of the `123 Patent as the corresponding structure for this limitation.

106. I agree with St. Lawrence that the disclosed corresponding function of this "means for" term is "amplifying the found codevector with said smoothing gain to thereby produce said gain-smoothed codevector." Further, with respect to this limitation, the `123 Patent states "[f]inally, the smoothed fixed codebook gain gs is calculated in gain smoothing calculator 228 … The smoothed gain gs is then used for scaling the innovative codevector ck in amplifier 232." Accordingly, I agree with St. Lawrence that the corresponding structure which performs this function is the amplifier (232) of Figure 2 of the `123 Patent.

107. Further, it was well-known to those of ordinary skill in the art in the relevant time frame that such an amplifier may be implemented through software based mathematical operations such as, for example, a multiplication. For example, the `123 Patent itself states that "to calculate the mean squared pitch prediction error for each vector $y^{(i)}$, the value $y^{(i)}$ is *multiplied* by the gain b by means of a corresponding *amplifier* $307^{(j)}$ and the value $by^{(j)}$ is subtracted from the target vector x by means of a corresponding subtractor $308^{(j)}$." (`123 Pat., 13:32-37). Additionally, the `123 Patent states that "[t]he generated scaled codevector gc_k *at the*

[8] Although my analysis focuses on independent claim 21, the same reasoning is applicable to the remaining claims in which this term is in dispute.

output of the amplifier 224 is processed through an innovation filter 205." (`123 Pat., 15:10-11). One of ordinary skill in the art in the relevant time frame would understand that "gc_k" refers to the gain "g" *multiplied* by the codevector "c_k." Accordingly, from the perspective of a person of ordinary skill in the art in the relevant time frame, the amplifier (232) was implemented using computer code.[9]

108. It is my understanding that defendants have not yet submitted an expert declaration regarding the issues discussed in this declaration. Further, it is my understanding that the defendants have not yet submitted a brief to the court explaining the basis for their positions with respect to the various claim terms. Accordingly, I reserve the right to supplement this declaration or otherwise address any arguments that may be raised by any expert retained by the defendants or by the defendants in their briefs.

I declare under penalty of perjury that the foregoing is true and correct.

____

TOKUNBO OGUNFUNMI

Executed on November 25, 2015

[9] *See also* my discussion in the "means for" section.